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Ramsden et al.

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(54) **METHOD FOR MEASURING PROCESSING DELAYS OF VOICE-OVER IP DEVICES**

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This patent is subject to a terminal disclaimer.

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Related U.S. Application Data

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H04M 3/22 (2006.01)
H04L 12/66 (2006.01)

(52) **U.S. Cl.**

CPC ***H04M 3/2227*** (2013.01); ***H04L 12/66*** (2013.01)

(58) **Field of Classification Search**

CPC ... H04W 24/00; H04L 43/50; H04L 43/0852; H04L 43/08; H04L 65/00; H04L 45/00; H04L 45/02; H04L 45/04; H04L 2012/64; H04L 12/64; H04L 65/102; H04L 29/06176; H04L 12/6418; H04B 17/003

See application file for complete search history.

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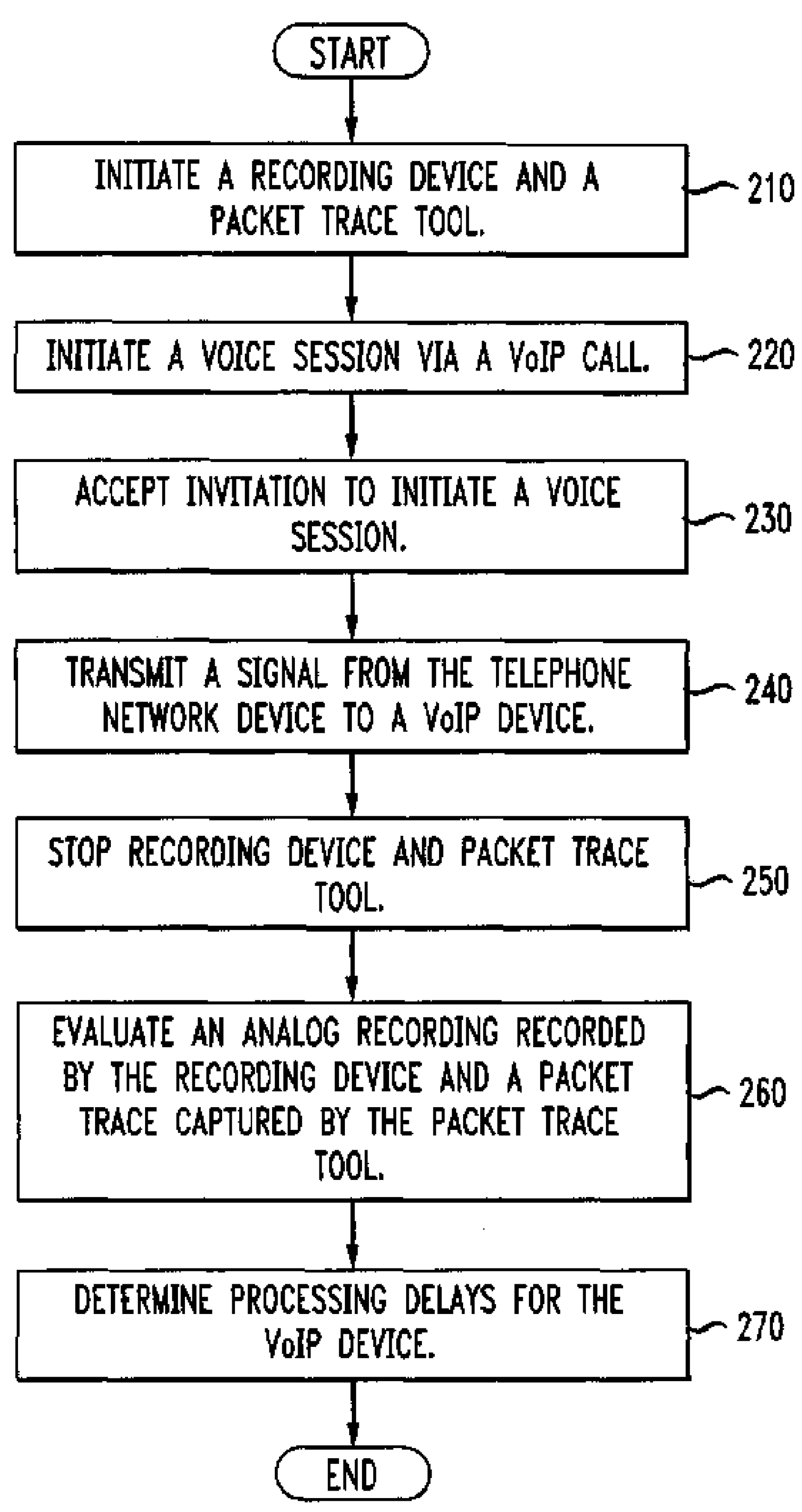
(57) **ABSTRACT**

A system and method for recording analog signals exchanged between a telephone device and a VoIP device, capturing packets exchanged between the VoIP device and an IP network, determining analog time values corresponding to analog characteristics of the analog signals, determining digital time values corresponding to digital characteristics of the packets, determining a common reference time for the analog time values and digital time values and determining a processing delay based on the analog time values and the digital time values.

20 Claims, 6 Drawing Sheets



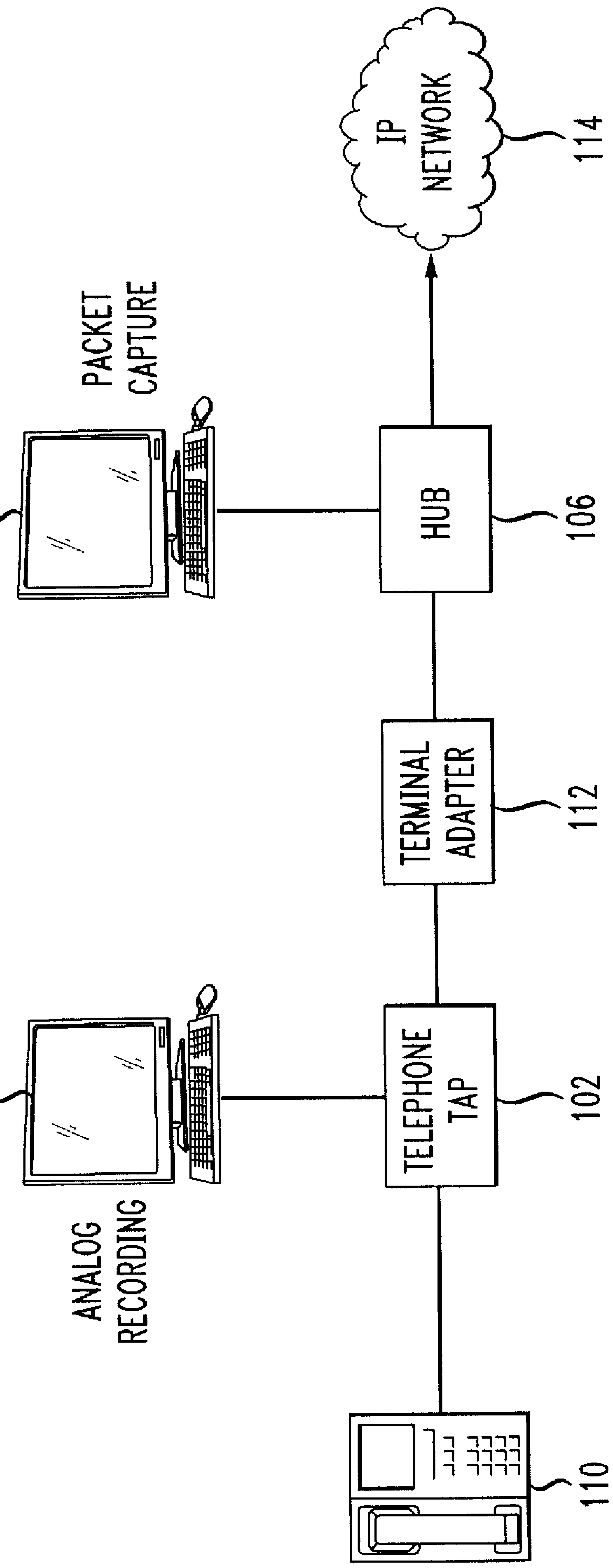
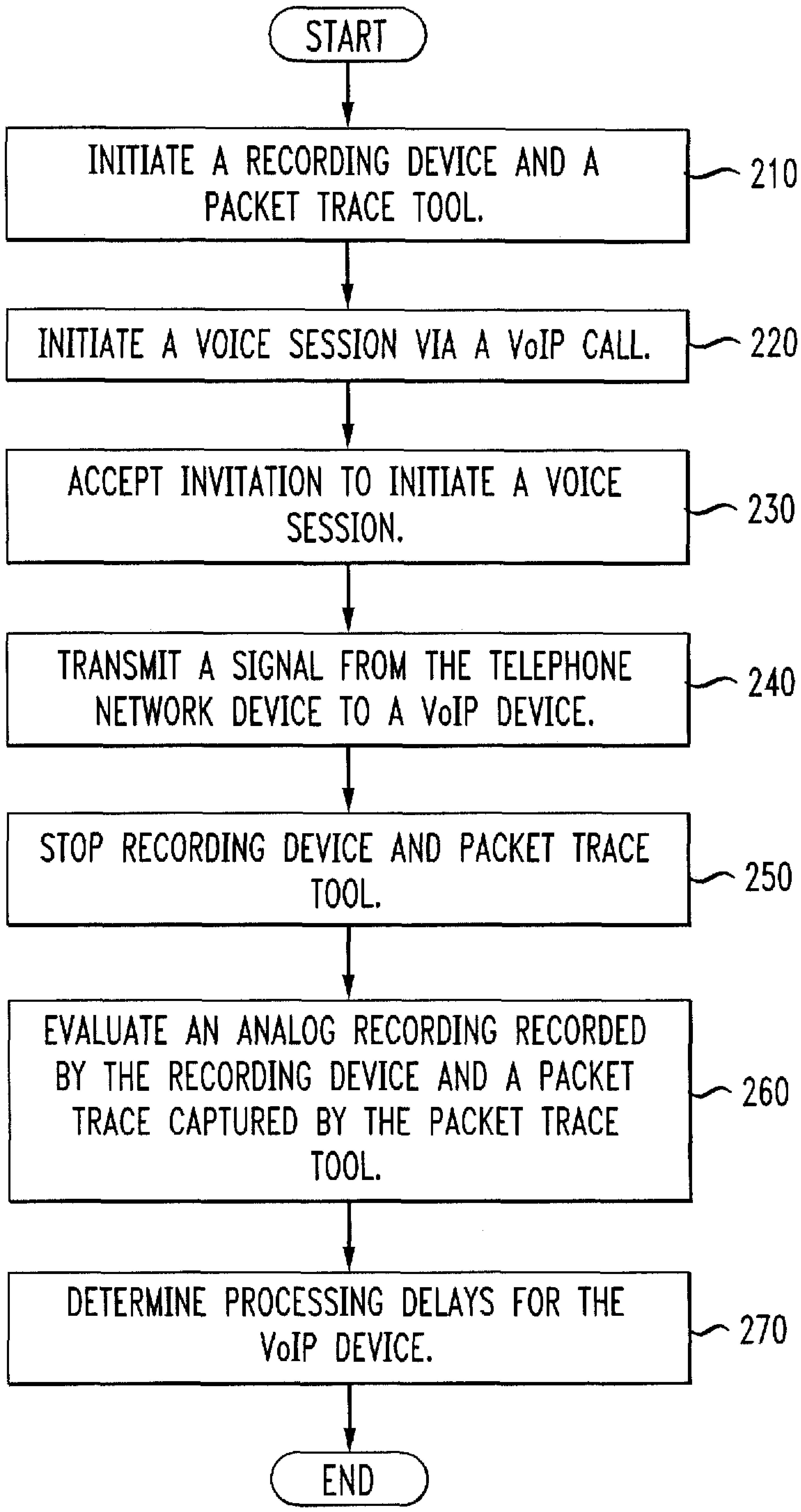
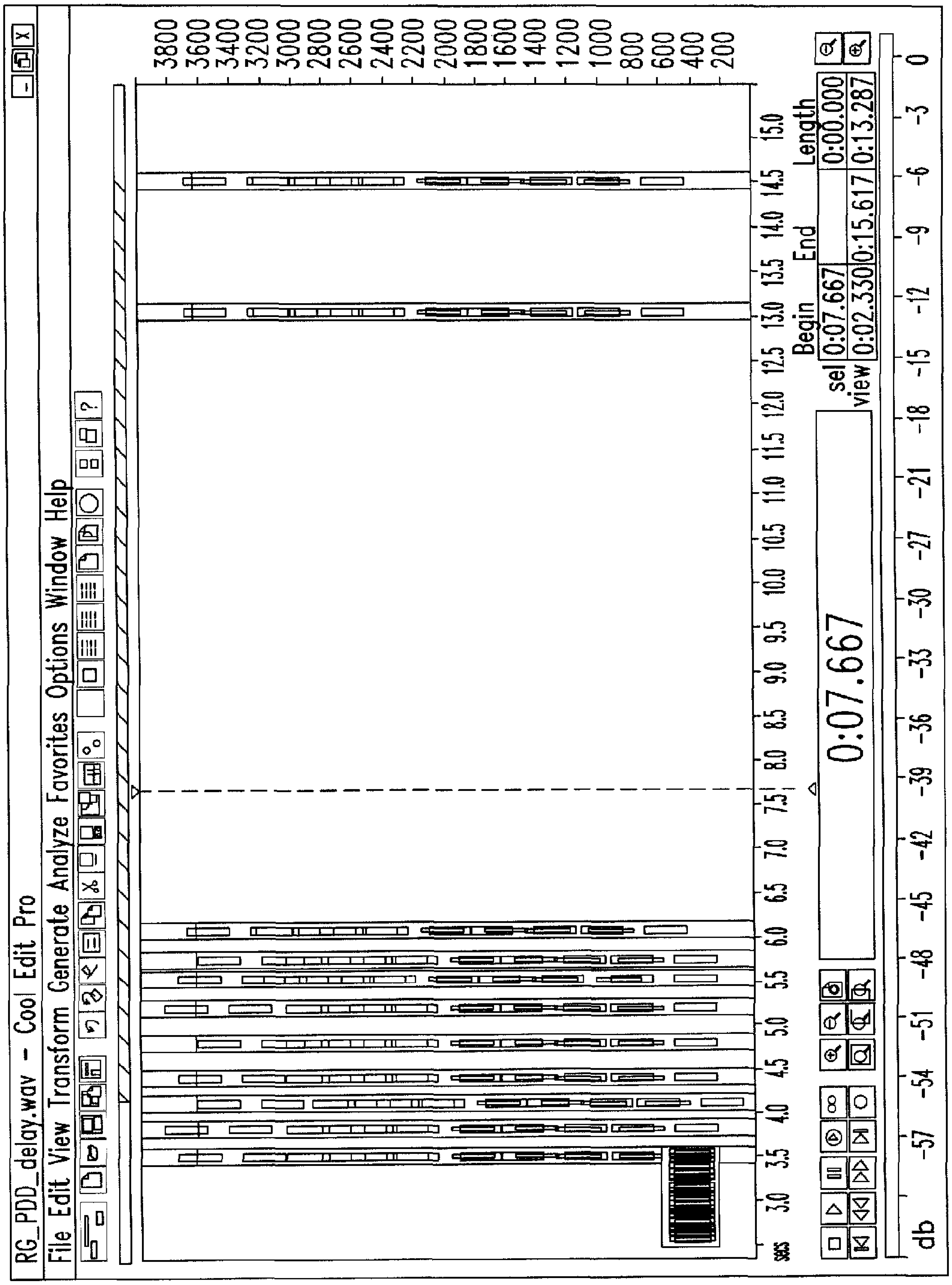

IP NETWORK
114
PACKET CAPTURE
108
HUB
106
TERMINAL ADAPTER
112
TELEPHONE TAP
102
104
ANALOG RECORDING
110

FIG. 1

FIG. 2


START
INITIATE A RECORDING DEVICE AND A PACKET TRACE TOOL.
210
INITIATE A VOICE SESSION VIA A VoIP CALL.
220
ACCEPT INVITATION TO INITIATE A VOICE SESSION.
230
TRANSMIT A SIGNAL FROM THE TELEPHONE NETWORK DEVICE TO A VoIP DEVICE.
240
STOP RECORDING DEVICE AND PACKET TRACE TOOL.
250
EVALUATE AN ANALOG RECORDING RECORDED BY THE RECORDING DEVICE AND A PACKET TRACE CAPTURED BY THE PACKET TRACE TOOL.
260
DETERMINE PROCESSING DELAYS FOR THE VoIP DEVICE.
270
END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File Edit View Transform Generate Analyze Favorites Options Window Help
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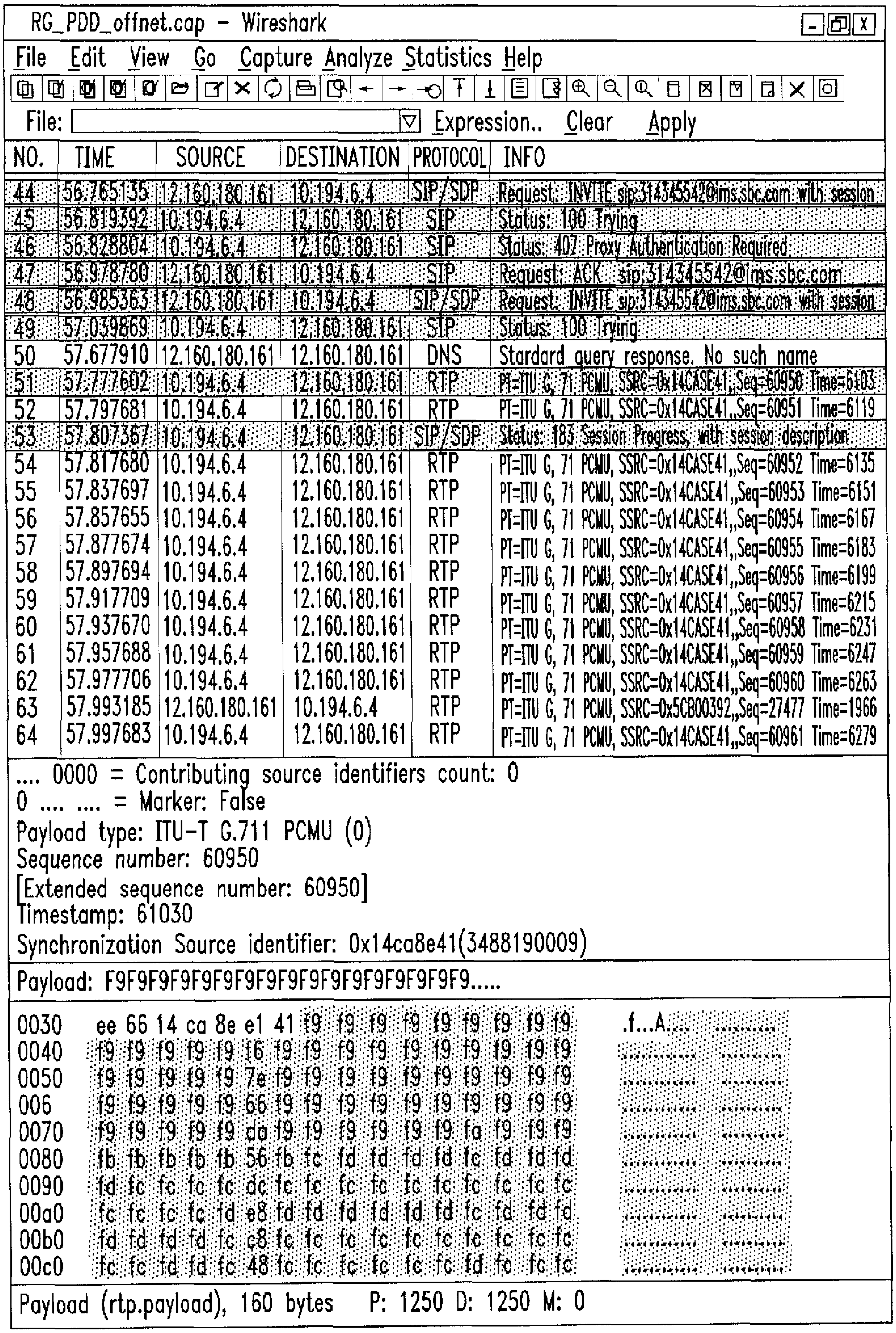
secs 3.0 3.5 4.0 4.5 5.0 5.5 6.0 6.5 7.0 7.5 8.0 8.5 9.0 9.5 10.0 10.5 11.0 11.5 12.0 12.5 13.0 13.5 14.0 14.5 15.0
0:07.667
Begin End Length
sel 0:07.667 0:00.000
view 0:02.330 0:15.617 0:13.287
db -57 -54 -51 -48 -45 -42 -39 -36 -33 -30 -27 -21 -18 -15 -12 -9 -6 -3 0

FIG. 3

FIG. 4A


RG_PDD_offnet.cap – Wireshark
File Edit View Go Capture Analyze Statistics Help
File: Expression.. Clear Apply
NO. TIME SOURCE DESTINATION PROTOCOL INFO
44 56.765135 12.160.180.161 10.194.6.4 SIP/SDP Request: INVITE sip:314345542@ims.sbc.com with session
45 56.819392 10.194.6.4 12.160.180.161 SIP Status: 100 Trying
46 56.828804 10.194.6.4 12.160.180.161 SIP Status: 407 Proxy Authentication Required
47 56.978780 12.160.180.161 10.194.6.4 SIP Request: ACK sip:314345542@ims.sbc.com
48 56.985363 12.160.180.161 10.194.6.4 SIP/SDP Request: INVITE sip:314345542@ims.sbc.com with session
49 57.039869 10.194.6.4 12.160.180.161 SIP Status: 100 Trying
50 57.677910 12.160.180.161 12.160.180.161 DNS Stardard query response. No such name
51 57.777602 10.194.6.4 12.160.180.161 RTP PT=ITU G, 71 PCMU, SSRC=0x14CASE41,,Seq=60950 Time=6103
52 57.797681 10.194.6.4 12.160.180.161 RTP PT=ITU G, 71 PCMU, SSRC=0x14CASE41,,Seq=60951 Time=6119
53 57.807367 10.194.6.4 12.160.180.161 SIP/SDP Status: 183 Session Progress, with session description
54 57.817680 10.194.6.4 12.160.180.161 RTP PT=ITU G, 71 PCMU, SSRC=0x14CASE41,,Seq=60952 Time=6135
55 57.837697 10.194.6.4 12.160.180.161 RTP PT=ITU G, 71 PCMU, SSRC=0x14CASE41,,Seq=60953 Time=6151
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57 57.877674 10.194.6.4 12.160.180.161 RTP PT=ITU G, 71 PCMU, SSRC=0x14CASE41,,Seq=60955 Time=6183
58 57.897694 10.194.6.4 12.160.180.161 RTP PT=ITU G, 71 PCMU, SSRC=0x14CASE41,,Seq=60956 Time=6199
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61 57.957688 10.194.6.4 12.160.180.161 RTP PT=ITU G, 71 PCMU, SSRC=0x14CASE41,,Seq=60959 Time=6247
62 57.977706 10.194.6.4 12.160.180.161 RTP PT=ITU G, 71 PCMU, SSRC=0x14CASE41,,Seq=60960 Time=6263
63 57.993185 12.160.180.161 10.194.6.4 RTP PT=ITU G, 71 PCMU, SSRC=0x5CB00392,,Seq=27477 Time=1966
64 57.997683 10.194.6.4 12.160.180.161 RTP PT=ITU G, 71 PCMU, SSRC=0x14CASE41,,Seq=60961 Time=6279
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0 .... = Marker: False
Payload type: ITU-T G.711 PCMU (0)
Sequence number: 60950
[Extended sequence number: 60950]
Timestamp: 61030
Synchronization Source identifier: 0x14ca8e41(3488190009)
Payload: F9F9F9F9F9F9F9F9F9F9F9F9F9F9F9F9.....
0030 ee 66 14 ca 8e e1 41 f9 f9 f9 f9 f9 f9 f9 f9 f9 .f...A...
0040 f9 f9 f9 f9 f9 f6 f9 f9 f9 f9 f9 f9 f9 f9 f9 f9
0050 f9 f9 f9 f9 f9 7e f9 f9 f9 f9 f9 f9 f9 f9 f9 f9
006 f9 f9 f9 f9 f9 66 f9 f9 f9 f9 f9 f9 f9 f9 f9 f9
0070 f9 f9 f9 f9 f9 da f9 f9 f9 f9 f9 f9 fa f9 f9 f9
0080 fb fb fb fb fb 56 fb fc fd fd fd fd fc fd fd fd
0090 fd fc fc fc fc dc fc fc fc fc fc fc fc fc fc fc
00a0 fc fc fc fc fd e8 fd fd fd fd fd fd fc fd fd fd
00b0 fd fd fd fd fc c8 fc fc fc fc fc fc fc fc fc fc
00c0 fc fc fd fd fc 48 fc fc fc fc fc fc fd fc fc fc
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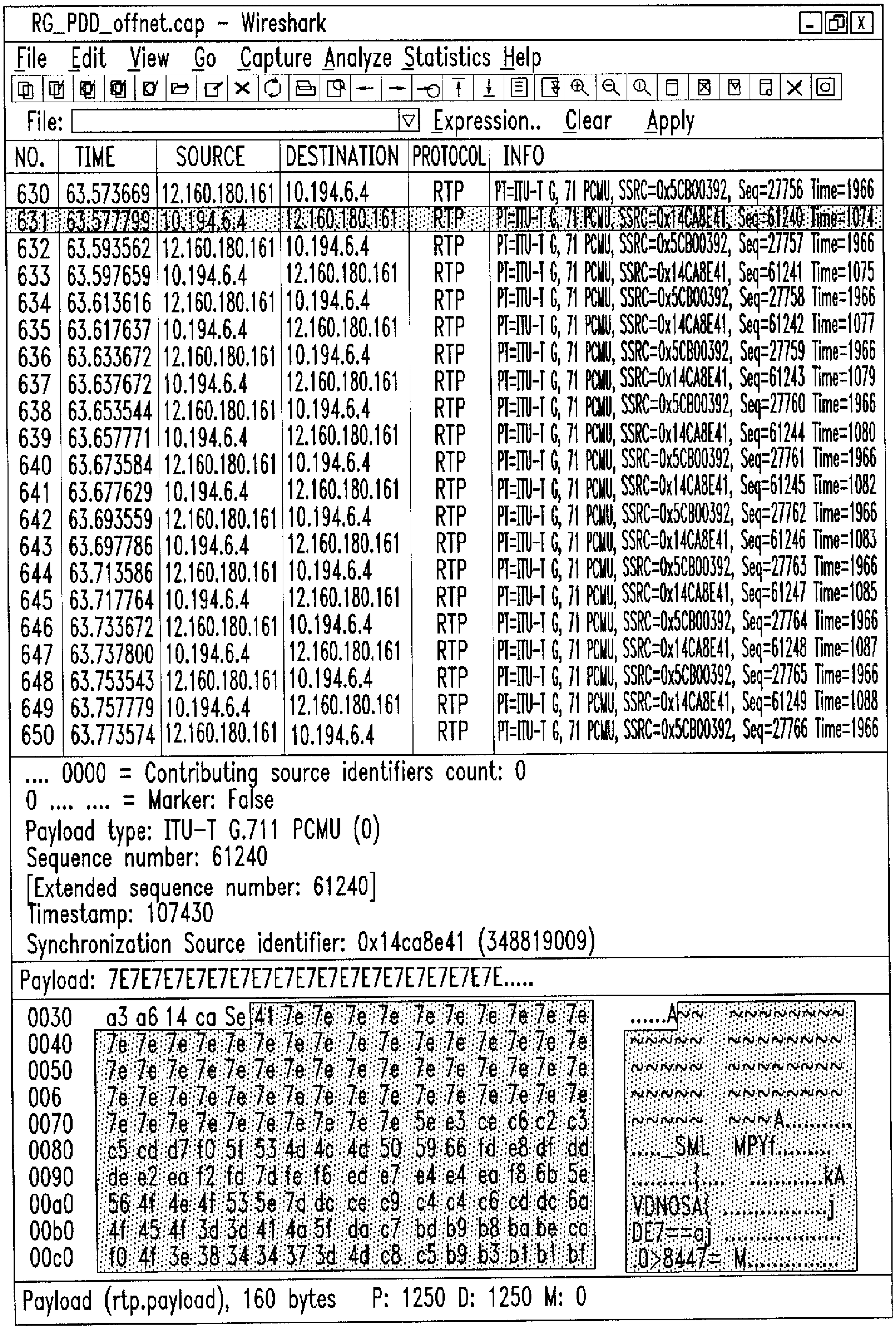
FIG. 4B


RG_PDD_offnet.cap – Wireshark
File Edit View Go Capture Analyze Statistics Help
File: Expression.. Clear Apply
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631 63.577799 10.194.6.4 12.160.180.161 RTP PT=ITU-T G, 71 PCMU, SSRC=0x14CA8E41, Seq=61240 Time=1074
632 63.593562 12.160.180.161 10.194.6.4 RTP PT=ITU-T G, 71 PCMU, SSRC=0x5CB00392, Seq=27757 Time=1966
633 63.597659 10.194.6.4 12.160.180.161 RTP PT=ITU-T G, 71 PCMU, SSRC=0x14CA8E41, Seq=61241 Time=1075
634 63.613616 12.160.180.161 10.194.6.4 RTP PT=ITU-T G, 71 PCMU, SSRC=0x5CB00392, Seq=27758 Time=1966
635 63.617637 10.194.6.4 12.160.180.161 RTP PT=ITU-T G, 71 PCMU, SSRC=0x14CA8E41, Seq=61242 Time=1077
636 63.633672 12.160.180.161 10.194.6.4 RTP PT=ITU-T G, 71 PCMU, SSRC=0x5CB00392, Seq=27759 Time=1966
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649 63.757779 10.194.6.4 12.160.180.161 RTP PT=ITU-T G, 71 PCMU, SSRC=0x14CA8E41, Seq=61249 Time=1088
650 63.773574 12.160.180.161 10.194.6.4 RTP PT=ITU-T G, 71 PCMU, SSRC=0x5CB00392, Seq=27766 Time=1966
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Timestamp: 107430
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Payload: 7E7E7E7E7E7E7E7E7E7E7E7E7E7E7E7E7E7E.....
0030 a3 a6 14 ca 5e 41 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e
0040 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e
0050 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e
006 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e
0070 7e 7e 7e 7e 7e 7e 7e 7e 7e 7e 5e e3 ce c6 c2 c3
0080 c5 cd d7 f0 5f 53 4d 4c 4d 50 59 66 fd e8 df dd
0090 de e2 ea f2 fd 7d fe f6 ed e7 e4 e4 ea f8 6b 5e
00a0 56 4f 4e 4f 53 5e 7d dc ce c9 c4 c4 c6 cd dc 6a
00b0 4f 45 4f 3d 3d 41 4a 5f da c7 bd b9 b8 ba be ca
00c0 f0 4f 3e 38 34 34 37 3d 4d c8 c5 b9 b3 b1 b1 bf
......A~~ ~~~~~~~~
~~~~~ ~~~~~~~~
~~~~~ ~~~~~~~~
~~~~~ ~~~~~~~~
~~~~~ ~~~A........
....._SML MPYf........
........}.... ....kA
VDNOSA{ ....j
DE7==aj ....
.0>8447= M..............
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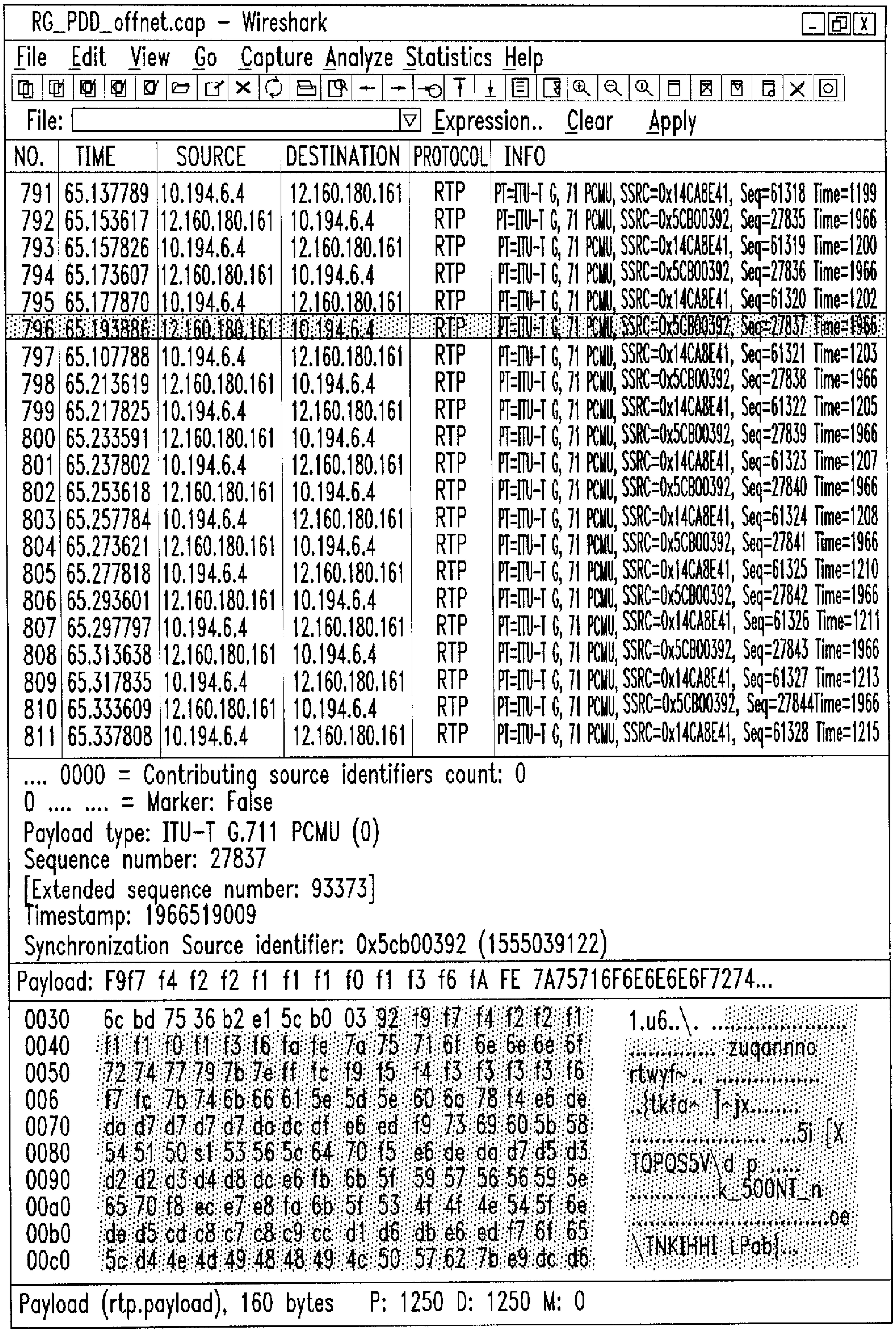
FIG. 4C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810 65.333609 12.160.180.161 10.194.6.4 RTP PT=ITU-T G, 71 PCMU, SSRC=0x5CB00392, Seq=27844Time=1966
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.... 0000 = Contributing source identifiers count: 0
0 .... = Marker: False
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Sequence number: 27837
[Extended sequence number: 93373]
Timestamp: 1966519009
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Payload: F9f7 f4 f2 f2 f1 f1 f1 f0 f1 f3 f6 fA FE 7A75716F6E6E6E6F7274...
0030 6c bd 75 36 b2 e1 5c b0 03 92 f9 f7 f4 f2 f2 f1
0040 f1 f1 f0 f1 f3 f6 fa fe 7a 75 71 6f 6e 6e 6e 6f
0050 72 74 77 79 7b 7e ff fc f9 f5 f4 f3 f3 f3 f3 f6
006 f7 fc 7b 74 6b 66 61 5e 5d 5e 60 6a 78 f4 e6 de
0070 da d7 d7 d7 d7 da dc df e6 ed f9 73 69 60 5b 58
0080 54 51 50 s1 53 56 5c 64 70 f5 e6 de da d7 d5 d3
0090 d2 d2 d3 d4 d8 dc e6 fb 6b 5f 59 57 56 56 59 5e
00a0 65 70 f8 ec e7 e8 fa 6b 5f 53 4f 4f 4e 54 5f 6e
00b0 de d5 cd c8 c7 c8 c9 cc d1 d6 db e6 ed f7 6f 65
00c0 5c d4 4e 4d 49 48 48 49 4c 50 57 62 7b e9 dc d6
Payload (rtp.payload), 160 bytes P: 1250 D: 1250 M: 0

METHOD FOR MEASURING PROCESSING DELAYS OF VOICE-OVER IP DEVICES

BACKGROUND

Voice-over Internet Protocol (VoIP) is a protocol optimized for the transmission of voice through the Internet or other packet-switched networks. VoIP allows users to use regular telephone networks anywhere through any Internet service provider. VoIP systems carry telephony signals as digital audio encapsulated in a data-packet stream over IP by converting analog audio signals to digital data that can be transmitted over the Internet. VoIP devices (e.g., terminal adapters, SIP phones, Media Gateways) are able to convert analog signals to digital signals encapsulated in the packets to be transmitted over the Internet.

VoIP may experience some time delays as a result of physical distance, the number of router hops, encryption and voice/data conversion. A typical packet will be forwarded over many links via many gateways, each of which will not begin to forward the packet until it has been completely received. Processing delays are incurred while a gateway determines what to do with a newly received packet. Small delays are generally not noticeable by users, but larger delays may result in a degradation of voice quality. Thus, it is desirable to measure processing delays associated with VoIP devices to determine whether processing delays added by VoIP devices are acceptable.

However, current measuring techniques use different tools to measure VoIP performance in the analog and IP domains, which makes it difficult to determine the time it takes for a VoIP device to convert an analog event into an IP event and vice versa. Thus, a method for accurately determining processing delays added by VoIP devices to end-to-end transmission delays and post-dialing delays may be beneficial to determine whether they are degrading the end-to-end performance of VoIP calls. Any company providing VoIP services or developing VoIP products may use the technique to assess the company's equipment performance.

SUMMARY OF THE INVENTION

A method for recording analog signals exchanged between a telephone device and a VoIP device, capturing packets exchanged between the VoIP device and an IP network, determining analog time values corresponding to analog characteristics of the analog signals, determining digital time values corresponding to digital characteristics of the packets, determining a common reference time for the analog time values and digital time values and determining a processing delay based on the analog time values and the digital time values.

A system measuring a processing delay of a VoIP device having a recording device recording analog signals exchanged between a telephone device and the VoIP device, a packet trace device capturing packets transmitted between the VoIP device and an IP network and a processor determining analog time values corresponding to analog characteristics of the analog signals, determining digital time values corresponding to digital characteristics of the packets, determining a common reference time for the analog time values and digital time values and determining a processing delay based on the analog time values and the digital time values.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** shows a system for measuring processing delays of VoIP devices according to an exemplary embodiment of the present invention.

FIG. **2** shows a method for measuring processing delays of VoIP devices according to an exemplary embodiment of the present invention.

FIG. **3** shows a screen view of an exemplary analog recording according to an exemplary system and method of the present invention.

FIG. **4**A shows a first screen view of an exemplary packet trace according to an exemplary system and method of the present invention.

FIG. **4**B shows a second screen view of the exemplary packet trace of FIG. **4**A.

FIG. **4**C shows a third screen view of the exemplary packet trace of FIG. **4**A.

DETAILED DESCRIPTION

The exemplary embodiments may be further understood with reference to the following description and the appended drawings, wherein like elements are referred to with the same reference numerals. The exemplary embodiments relate to a system and method for measuring processing delays associated with VoIP devices. It should be noted that although exemplary embodiments of the present invention show and/or describe measuring the processing delay of a terminal adapter, the system and method of the present invention may be used to measure processing delays of any VoIP device such as, for example, SIP phones, Media Gateways, etc.

As shown in FIG. **1**, a system **100** according to an exemplary embodiment of the present invention comprises a recording device **104** and a packet trace tool **108**. The recording device **104** is connectable between a telephone network device **110** and a VoIP device **112** via a telephone tap **102** such that the recording device **104** may record an analog signal that is transmitted between the telephone network device **110** and the VoIP device **112**. The packet trace tool **108** is connectable between the VoIP device **112** and an IP Network **114** via a hub **106** such that the packet trace tool **108** may take a packet capture of packets transmitted between the VoIP device **112** and the IP Network **114**. In the example shown, the telephone network device **110** is a standard telephone while the VoIP device **112** is a terminal adapter. It will be understood by those of skill in the art, however, that any telephone network device **110** and VoIP devices **112** may be used in the present invention. For example, the telephone network device **110** may also be a network switch, a PBX or other similar device. The VoIP device **112** may also be a SIP phone, a Media Gateway or other similar device.

The recording device **104** may be a personal computer, server or other processing arrangement loaded with an audio recording software such as Cool Edit. Once initiated, the recording device **104** records the analog signal transmitted between the telephone network device **110** and the VoIP device **112**. The recording device **104** may record the analog signal as a digital soundwave. The recording device **104** is able to access the signal to/from the telephone network device **110** via the telephone tap **102**, which monitors the connection between the telephone network device **110** and the VoIP device **112**. It will be understood by those of skill in the art that the recording device **104** may include a memory, a display and/or an audio playback device such that the recordings of the analog signal may be stored and/or displayed or played back by a user.

The packet trace tool **108** may be a personal computer, a server or other processing arrangement that is loaded with a packet analyzer such as Wireshark. The packet trace tool **108** allows a user to see all traffic being passed to/from the IP network **114** such that packets being transmitted therebetween may be recorded in a packet capture. As data streams flow across the network **114**, the packet trace tool **108** captures each packet and eventually decodes and analyzes its contents. The packet trace tool **108** is able to access the packets passing between the VoIP device **112** and the IP Network **114** via the hub **106**, which permits connection of the packet trace tool **108** to both the VoIP device **112** and the IP network **114**.

Although the recording device **104** and the packet trace tool **108** are shown as two separate devices, it will understood by those of skill in the art that the recording device **104** and the packet trace tool **108** may be a single device. For example, the recording device **104** and the packet trace tool **108** may be a single personal computer, server or other processing arrangement loaded with both the audio recording software and the packet analysis tool. It will also be understood by those of skill in the art that the recorded analog signals and the recorded packet capture may be transmitted to a processor (not shown) for determining time values that may be used to calculate a processing delay. The processor may be a separate personal computer, server or other processing arrangement connected to both the recording device **104** and the processing tool **108** such that the recorded analog signal and the recorded packet capture may be analyzed. Alternatively, the processor may be a component of one of the recording device **104** and/or the packet trace tool **108**.

FIG. **2** shows an exemplary method **200** according to the present invention, using the system **100**, as described above. The method **200** comprises a first step **210** in which the recording device **104** and the packet capture tool **108** are initiated. Upon initiation, the recording device **104** will record all analog signals transmitted between the telephone network device **110** and the VoIP device **112** in an analog recording and the packet trace tool **108** will capture all the data transmitted between the VoIP device **112** and the IP Network **114** in a packet capture. Once the recording device **104** and the packet capture tool **108** have been initiated, a telephone call may be initiated by a first user via the telephone network device **110** (e.g., a telephone) in a step **220**. The telephone call is initiated by dialing digits on the telephone network device **110**. These dialed digits may be converted to digital packets that are sent via a signaling protocol such as Session Initiation Protocol (SIP). A SIP Invite is sent to a second user, via the IP Network **114**, at the telephone number being dialed to invite the second user to establish a voice session. The SIP Invite is transmitted across the network and causes a ringing signal to be sent to the second user. Once ringing begins at a telephone network device of the second user, a digital packet of the ringing signal is transmitted back to the VoIP device **112** to be transmitted to the telephone network device **110** so that the first user may also hear a ringing signal, which would indicate that the call has been connected.

When ringing is heard by the second user, the second user may accept the first user's invitation for a voice session, in a step **230**. The invitation may be accepted by answering the phone call upon hearing the phone ringing. Once the voice session has been established, a speech signal may be sent from the telephone network device **110** to the VoIP device **112**, in a step **240**. The speech signal may be a tone, speech or any other analog signal sent by the first user via the telephone network device **110**. The VoIP device **112** then converts the analog speech signal to a packet that may be transmitted using a format such as Real-time Transfer Protocol (RTP). The speech signal may eventually be used to relate times that are recorded by the recording device **104** and the packet trace tool **108**. Once the speech signal has been transmitted, the recording device **104** and the packet trace tool **108** may be stopped, in a step **250**.

It will be understood by those of skill in the art that the analog recording of the recording device **104** and the packet capture of the packet tracing tool **108** may be saved to a memory of each of the devices **104**, **108** and/or displayed on a display of each of the devices **104**, **108**. Alternatively, the analog recoding and/or the packet capture may be stored and/or displayed on a separate memory and/or display.

In a step **260**, the analog recordings of the recording device **104** are evaluated to identify a first time T_1, a second time T_2 and a third time T_3 and the packet trace is evaluated to determined a first time S_1, a second time S_2 and a third time S_3. The times T_1, T_2 and T_3 may be referred to herein as "analog time values" to indicate the times have been derived from the analog signal. Those skilled in the art will understand that other time values may also be derived from the analog signal and these would also be referred to as analog time values. Similarly, S_1, S_2 and S_3 may be referred to as "digital time values" to indicate the times have been derived from the packet trace. For the analog recordings, the first time T_1 corresponds to a time in which a final Dual Tone Multi-Frequency (DTMF) digit ended, in other words, when the last digit of the telephone number that is being dialed has been entered. The second time T_2 corresponds to a time when the ringing signal starts. The third time T_3 corresponds to a time when the transmitted signal starts. For the packet trace, the first time S_1 corresponds to a time that the SIP Invite was sent. The second time S_2 corresponds to a time when the ringing signal starts. The RTP packets transmitted from the IP Network **114** to the VoIP device **112** may be evaluated to find when the ringing signal starts. The third time S_3 corresponds to the time when the transmitted signal starts. The RTP packets transmitted from the VoIP device **112** to the IP Network **114** may be evaluated to find when the speech signal starts.

In a step **270**, the time values identified in the step **260** may then be used to determine a processing time of the VoIP device **112** to create the SIP Invite packet, D_{SIP}, and a processing time that it takes the VoIP device **112** to convert an RTP packet to an analog signal that is transmitted to the telephone network device **110** in the step **240**, D_{RTPRCV}. The processing delays D_{SIP} and D_{RTPRCV} may be determined by relating the time values T_1, T_2, T_3 of the analog recording to the time values S_1, S_2, S_3 of the packet capture. The analog recording and the packet trace may be related using the following equations:

$$T_1+D_{SIP}=S_1+\Delta \tag{1}$$

$$T_2=S_2+D_{RTPRCV}+\Delta \tag{2}$$

$$T_3+D_{RTPXMT}=S_3+\Delta \tag{3}$$

Where, Δ is a difference between when the analog recording and packet captures were started, D_{SIP} is the processing time used by the VoIP device **112** to create the SIP Invite packet, D_{RTPXMT} is a processing time it takes the VoIP device **112** to convert the speech signal received from the phone into the RTP packet and D_{RTPRCV} is the processing time the VoIP device **112** takes to convert the RTP packet into the analog signal that is transmitted.

Using the values of T_1, T_2, T_3, S_1, 5_2 and S_3 that are identified by evaluating the analog recording and the packet capture in the equations disclosed above, processing delays of the VoIP device **112** are calculated in the step **270**. Specifically, it is possible to calculate D_{RTPRCV}, the time it takes the VoIP device **112** to convert an RTP packet into the analog signal, and D_{SIP}, the time it takes the VoIP device **112** to create the SIP Invite packet. The value of D_{RTPXMT} may be easily determined within a few ms. For example, if the VoIP device **112** of the system **100** is a terminal adapter that is using 20 ms RTP packets and a G.711 codec, D_{RTPXMT} may be estimated to be approximately 21 ms. Additionally, Δ may be easily eliminated by taking a difference between the first equation and the third equation and a difference between the second equation and the third equation, since the value of D_{RTPXMT}, of the third equation, is known.

The difference of the first equation and the third equation would result in an equation that would determine a value of D_{SIP} as follows:

$$D_{SIP}=(T_3-T_1)-(S_3-S_1)+21\text{ ms.}$$

The difference of the second equation and the third equation would result in an equation that would determine a value of D_{RTPRCV} as follows:

$$D_{RTPRCV}=(S_3-S_2)-(T_3-T_2)-21\text{ ms.}$$

Thus, using these equations, the time it takes the VoIP device **112** to convert an RTP packet into the analog signal, D_{RTPRCV}, and the time it takes the VoIP device. **112** to create the SIP Invite packet, D_{SIP}, may be determined accordingly.

For example, FIG. **3** shows a screenshot of an exemplary analog recording according to the present invention. In the example shown, the last DTMF digit was dialed at approximately **6** seconds, the ringing signal starts at approximately **13** seconds and the transmitted signal starts at approximately **14**.**4** seconds. Since the equations above are measures in milliseconds, the values of T_1=6000 ms, T_2=13000 ms and T_3=14200 ms may be used in the above equations.

FIGS. **4**A-**4**C show a series of screenshots of an exemplary packet capture using Wireshark. As shown in FIG. **4**A, packet no. 44, a SIP Invite is sent from the telephone network device **110** to the VoIP device **112** at approximately 56.8 seconds. Thus, S_1=56800 ms. The values of S_2 and S_3 may be determined by evaluating the payloads of the transmitted and received RTP packets in the capture. When no signal is being transmitted, the payload shows a repetitive sequence of hexadecimal values, which indicates that the channel is idle or that there is a low level noise present in the channel. As shown in FIG. **4**B, the payload begins to transition from a repeating pattern to a varying pattern, indicating that a signal is being transmitted. Based on this assessment of the payload, it can be determined that packet no. 631 shows the first RTP packet of the ringing signal that is sent from the VoIP device **112** to the telephone network device **110** at approximately 63.6 seconds. Evaluating the payload of FIG. **4**C in a similar manner, it may be determined that packet no. 796 shows the RTP packet being transmitted from the telephone network device **110** to the VoIP device **112** at approximately 65.2 seconds. Thus, the values of S_2=63600 ms and S_3=65200 ms may be used in the equations above. Alternatively, it will be understood by those of skill in the art that the values of S_2 and S_3 may be determined by converting the RTP packets' payloads to an audio file and using a recording device such as Cooledit to determine when audio begins in each direction.

Using the exemplary time values of FIGS. **3** and **4**A-**4**C we may calculate D_{SIP} and D_{RTPXMT} as follows:

$$D_{SIP}=(T_3-T_1)-(S_3-S_1)+21\text{ ms}$$

$$D_{SIP}=(14400\text{ ms}-6000\text{ ms})-(65200\text{ ms}-56800\text{ ms})+21\text{ ms}$$

$$D_{SIP}=8400\text{ ms}-8400\text{ ms}+21\text{ ms}$$

$$D_{SIP}=21\text{ ms}$$

$$D_{RTPRCV}=(S_3-S_2)-(T_3-T_2)-21\text{ ms}$$

$$D_{RTPRCV}=(65200\text{ ms}-63600\text{ ms})-(14400\text{ ms}-13000\text{ ms})-21\text{ ms}$$

$$D_{RTPRCV}=1600\text{ ms}-1400\text{ ms}-21\text{ ms}$$

$$D_{RTPRCV}=179\text{ ms}$$

Thus, based on the exemplary values determined by evaluating the packet capture and the analog recording, the values of D_{SIP}, the processing time used by the VoIP device **112** to create the SIP Invite and the D_{RTPXMT}, the processing time it takes to convert the signal received from the telephone network device **110** to the RTP packet, may be determined. It should be noted, however, that the above values are exemplary values only.

As can be seen from the above examples, the exemplary embodiments allow for the use of separate capture devices in the analog domain (e.g., the analog recording device) and the IP domain (e.g., the packet capture device) to take advantage of the functionality of these devices. However, the exemplary embodiments call for a common reference time for a signal that is measured by both devices to create a relationship in the domain between the measurements of both devices, thereby allowing overall processing times to be related and determined.

It will be apparent to those skilled in the art that various modifications and variations may be made in the structure and methodology of the present invention, without departing from the spirit or scope of the invention. Thus, it is intended that the present invention cover the modifications and variations of this invention provided that they come within the scope of the appended claims and their equivalents.

What is claimed is:

1. A method, comprising:

recording analog signals exchanged between a telephone device and a Voice-over Internet Protocol ("VoIP") terminal adapter;

capturing packets exchanged between the VoIP terminal adapter and an Internet Protocol ("IP") network;

determining analog time values corresponding to analog characteristics of the analog signals;

determining digital time values corresponding to digital characteristics of the packets, wherein the digital characteristics include a Session Initiation Protocol ("SIP") invite packet;

determining a common reference time for the analog time values and digital time values; and

determining a processing delay based on a difference between an analog time difference and a digital time difference.

2. The method of claim **1**, wherein the analog time difference is determined based on T_3-T_1, wherein T_1 is a first analog time value of a first analog characteristic corresponding to when a final digit is dialed from the telephone device and T_3 is a second analog time value of a second analog characteristic corresponding to when a speech signal starts.

3. The method of claim **1**, wherein the digital time difference is determined based on S_3-S_1, wherein S_1 is a first digital time value of a first digital characteristic corresponding to when the SIP invite packet was sent and S_3 is a second digital time value of a second digital characteristic corresponding to when a speech signal starts.

4. The method of claim **1**, wherein the analog characteristics include one of a final Dual Tone Multi-Frequency digit of a dialed telephone number, a start of a ringing and a start of a transmitted signal.

5. The method of claim **1**, wherein the digital characteristics further include a start of a ringing and a start of a transmitted signal.

6. The method of claim **1**, wherein the processing delay is determined by adding a processing time of the VoIP terminal adapter to convert an analog signal into a digital signal to the difference between the analog time difference and the digital time difference.

7. The method of claim **1**, wherein the processing delay is a processing time of the VoIP terminal adapter to create the SIP invite packet.

8. The method of claim **1**, wherein the processing delay is a processing time of the VoIP terminal adapter to convert a packet into an analog signal.

9. The method of claim **1**, further comprising:
stopping the recording and capturing after a speech signal has been transmitted from the telephone device to the VoIP terminal adapter.

10. The method of claim **1**, further comprising:
storing at least one of the analog signals and the packets in a memory.

11. A system, comprising:
a recording device recording analog signals exchanged between a telephone device and a Voice-over Internet Protocol ("VoIP") terminal adapter;
a packet trace device capturing packets transmitted between the VoIP terminal adapter and an Internet Protocol ("IP") network; and
a processor determining analog time values corresponding to analog characteristics of the analog signals, determining digital time values corresponding to digital characteristics of the packets, wherein the digital characteristics include a Session Initiation Protocol ("SIP") invite packet, determining a common reference time for the analog time values and digital time values and determining a processing delay based on a difference between an analog time difference and a digital time difference.

12. The system of claim **11**, wherein the analog time difference is determined based on T_3-T_1, wherein T_1 is a first analog time value of a first analog characteristic corresponding to when a final digit is dialed from the telephone device and T_3 is a second analog time value of a second analog characteristic corresponding to when a speech signal starts.

13. The system of claim **11**, wherein the digital time difference is determined lined based on S_3-S_1, wherein S_1 is a first digital time value of a first digital characteristic corresponding to when the SIP invite packet was sent and S_3 is a second digital time value of a second digital characteristic corresponding to when a speech signal starts.

14. The system of claim **11**, wherein the analog characteristics include one of a final Dual Tone Multi-Frequency digit of a dialed telephone number, a start of a ringing and a start of a transmitted signal.

15. The system of claim **11**, wherein the digital characteristics further include a start of a ringing and a start of a transmitted signal.

16. The system of claim **11**, wherein the processing delay is determined by adding a processing time of the VoIP terminal adapter to convert an analog signal into a digital signal to the difference between the analog time difference and the digital time difference.

17. The system of claim **11**, wherein the processing delay is a processing time of the VoIP terminal adapter to create a SIP invite packet.

18. The system of claim **11**, wherein the processing delay is a processing time of the VoIP terminal adapter to convert a packet into an analog signal.

19. A non-transitory computer-readable storage medium including a set of instructions executable by a processor, the set of instructions, when executed by the processor, causing the processor to perform operations comprising:
recording analog signals exchanged between a telephone device and a Voice-over Internet Protocol ("VoIP") terminal adapter;
capturing packets exchanged between the VoIP terminal adapter and an Internet Protocl ("IP") network;
determining analog time values corresponding to analog characteristics of the analog signals;
determining digital time values corresponding to digital characteristics of the packets, wherein the digital characteristics include a Session Initiation Protocol ("SIP") invite;
determining a common reference time for the analog time values and digital time values; and
determining a processing time of the VoIP terminal adapter to create a SIP invite packet based on a difference between an analog time difference and a digital time difference.

20. The non-transitory computer-readable storage medium of claim **19**, wherein the analog time difference is determined based on T_3-T_1, wherein T_1 is a first analog time value of a first analog characteristic corresponding to when a final digit is dialed from the telephone device and T_3 is a second analog time value of a second analog characteristic corresponding to when a speech signal starts, and wherein the digital time difference is determined based on S_3-S_1, wherein S_1 is a first digital time value of a first digital characteristic corresponding to when the SIP invite packet was sent and S_3 is a second digital time value of a second digital characteristic corresponding to when the speech signal starts.

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